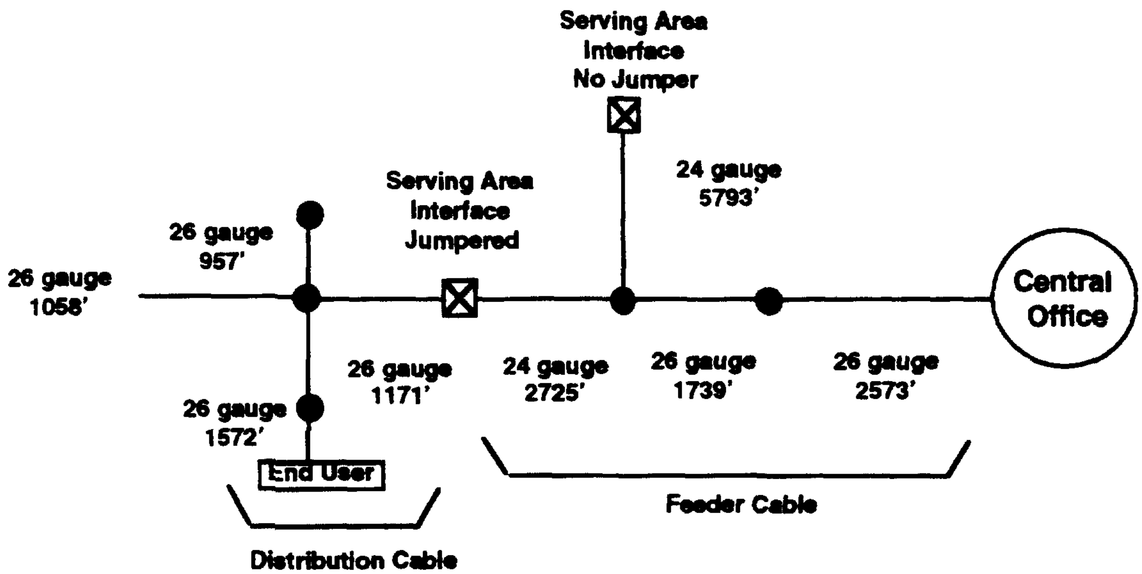

Figure 4: Cable Makeup of a Typical Metallic Loop

Even though V.34 modems have been specifically designed to overcome the effects of most analog impairments, loop attenuation at the higher voice band frequencies can still represent a prominent roadblock to achieving high connect rates. Figure 5 illustrates the attenuation per mile of a 26 gauge non-loaded cable pair. Note that the loss per mile at the V.34 upper nyquist frequency of approximately 3700 Hz is over 5 dB per mile.

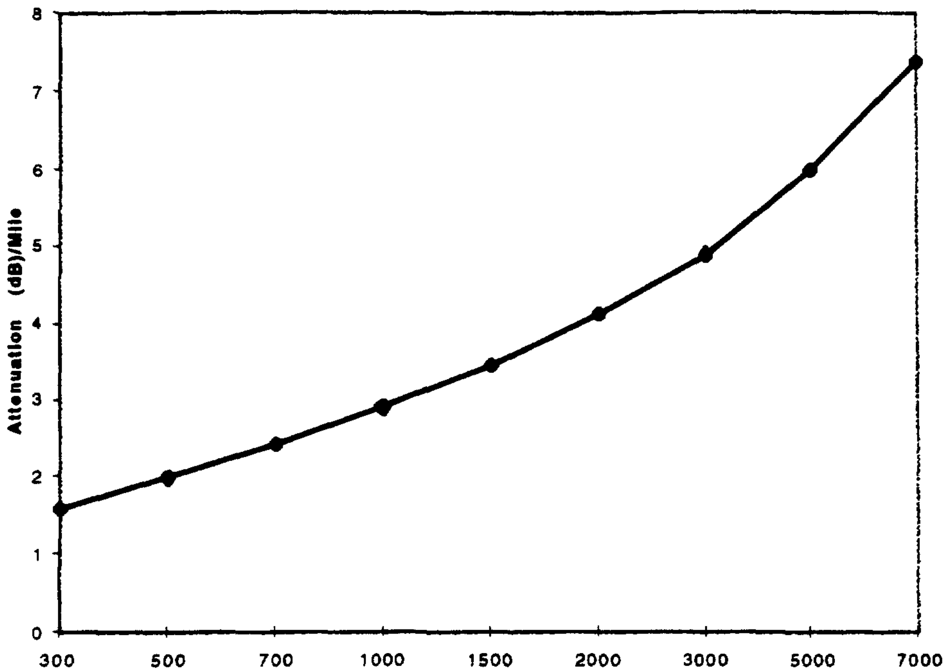

Figure 5: Attenuation/Mile (dB) of 26 AWG N.L. Cable @ 70 Degrees F

The noise level limit for a local loop is specified by Bellcore at 21 dBrnC0. Under normal conditions, the SNR of an end-to-end connection is not controlled by loop noise.

It should be emphasized that loops can vary dramatically by age and general condition of the plant. Older plant may have poor splices or suffer from water leakage which will change its electrical characteristics. Two analog loops can be involved in a connection and their affects are cumulative One end in isolation may not be the root cause of a performance problem but, when taken together, could prevent high DCE connect rates. Eliminating at least one analog loop in a connection will improve V.34 performance.

5.1.2 Filters in Channel Units/Line Cards

Telephone companies couple the local loop to the network through line transformers and filters that limit the bandwidth of the voice band channel. A typical filter design specification, showing upper and lower attenuation bounds, for a line card is indicated in Figure 6. It should be understood that there is nothing inherently band limiting in the copper wires themselves. Technologies such as Asymmetric Digital Subscriber Line, for example, are specifically designed for non-loaded loops that are not bandwidth limited and use frequencies up to 1.5 MHz. The logical question then is why employ filters? The answer is both historical and practical.

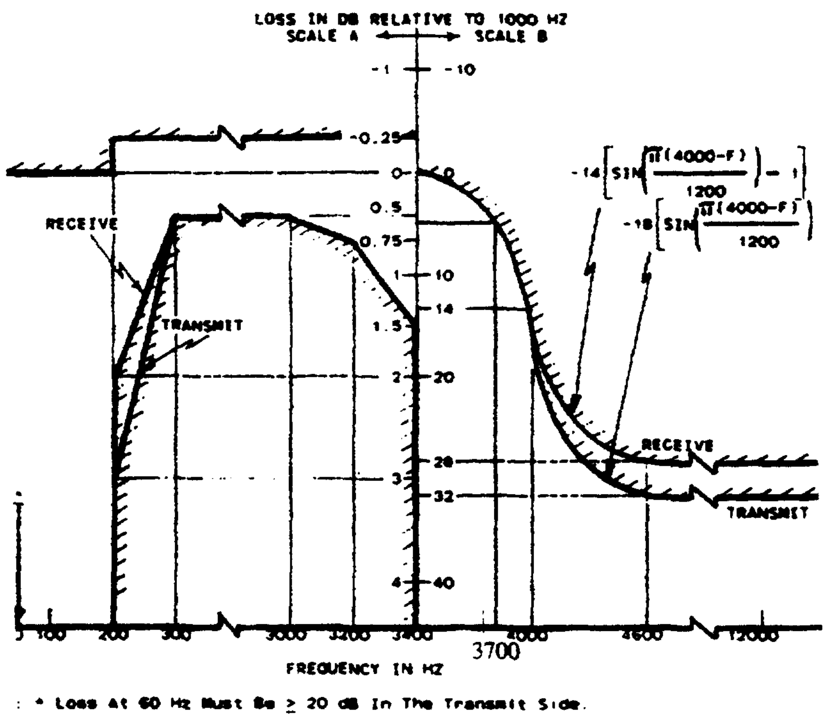

Figure 6: Transfer Function Bounds for a Typical Line Card Filter Design

Filtering is necessary to set limits on the range over which devices can practically operate and to limit energy from folding back into the passband after being sampled at 8 kHz. In addition, there are limits to a system's need to operate above a useful frequency range especially if no new information is conveyed. In the case of speech, early "grade of service" experiments yielded results indicating that a range of frequencies between 200 and 3400 Hz yielded sufficient voice quality to carry on normal conversation without loss of intelligibility. Increasing the frequency range of the network would have also meant increasing the cost. Thus, practical limits were set to minimize network cost without sacrificing voice quality. Digital systems permit a bandwidth of up to 4000 Hz. The PCM process in digital telephone systems sample at twice this frequency (8 kHz). An 8 kHz sampling rate requires the use of anti-aliasing filters to effectively remove high frequency components above 4 kHz.

The upper Nyquist frequency for the highest symbol rate specified in the V.34 standard is 3673 Hz. Since V.34 modems require detectable energy at these higher frequencies to operate at their maximum rate, attenuation should be calculated at these higher frequencies (versus 1004 Hz for conventional voice telephony). Losses at this frequency (3700 Hz) for most CODEC filters in use in the PSTN are at least 4 dB as shown in Figure 6. (It is believed that typical transmit and receive filter designs for the region above 3400 Hz provide attenuation close to the upper bound in Figure 5).

Table 2 in section 4.6 basically illustrates two means for increasing modem transmission rate: 1.) improve the SNR, and/or 2.) increase the bandwidth of the channel[4].

Improving the SNR on a copper pair can be achieved either by increasing the signal amplitude or by reducing the noise component. Signal amplitude must meet FCC regulations for an RJ-11 (permissive voice jack). Noise levels can be reduced by better plant construction, shielding and eliminating noise sources. All of these are possible in the telephone network if a given circuit is "designed". POTS and Measured Business lines use RJ-11 interfaces and are therefore not designed. It is therefore not possible to achieve 28.8kb/s on every local loop.

5.1.3 Load Coils

Load coils are installed on long loops to reduce attenuation distortion. Attenuation distortion is caused by the resistance and capacitance of the cable pair. This distortion causes high frequency components in the voice signal to be attenuated more than low frequencies causing poor audio quality. To solve this problem, load coils are used to "flatten" out the attenuation across the passband. Unfortunately, the end result is a sharper roll off or filtering effect at the high frequency end of the passband. Figure 7 shows the band edge filtering effect of a loaded cable compared with that of a typical line card filter. Note that the loss at 3700 Hz for a loop with 18 kft of 24 gauge wire is about 15 dB, 10 dB higher than a line card filter. Load coils could cause the modem to fall back to a lower symbol rate.

[4] Recall that baud is related to bandwidth by Nyquist, where Baud=2W where W is the bandwidth of the channel.

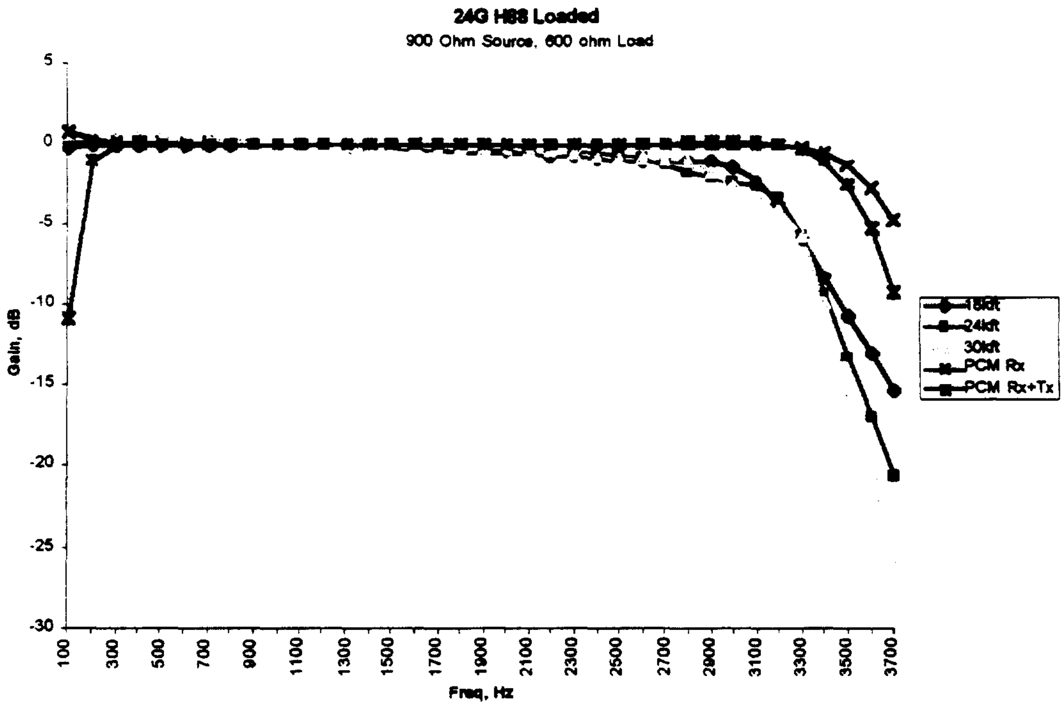

Figure 7: Transfer Function of PCM Filters and Loaded 24 Gauge Cable

5.2 Signal to Noise Level Impairments

Digital impairments in the network can also cause users to experience lower throughput. Digital impairments are a result of the Pulse Code Modulation (PCM) process and the Digital Network Loss Plan.

Digital impairments that affect modems result from CODECs (which include 2-4 Wire Conversion Hybrids), digital loss pads and in-band signaling on digital telephone transmission equipment. Table 4 summarizes these impairments relative to SNR degradation in dB; each impairment is discussed later in this section. The table is divided into two sections. The heavily outlined upper portion of the table contains digital impairments known to affect modem performance. The impairments listed in the second portion, outlined in a heavy dashed line, can cause problems but usually do not. The last column describes steps that can be taken to either eliminate or minimize the effect of an impairment.

Table 4: Summary of PSTN SNR Impairments

Network Equipment	Network Location	*Equiv SNR Loss/ Occurrence	Impairment(s)	Some Remedies
CODECS Linked for Digital Line Cards that terminate 2-wire circuits	Channel Banks DLC Remote Terminals DLC CO Terminals Digital Switch LUs	3dB 2nd 1.8dB 3rd 1.2 dB 4th etc.	Introduces quantization noise in the signal for each PCM link.	Minimize D/A conversions in circuit designs
HYBRIDS	Employed along with CODECS in all conversions to 2-wire loops	Variable	Echo from 4-2wire conversions. Envelope Delay Distortion.	Minimize D/A Conversions. Use Dynamic balance network that compensates for differing loop characteristics
Robbed Bit Signaling	Channel Banks Remote Terminals TR-08 TR-303	1.8-2.5 dB 1st 1.2-1.5 dB 2nd	Robs least significant bit of every sixth frame. Increases quanitzing noise. Depends on DS1 Superframe alignment	Use out-of-band signaling.
Super Frame Alignment	Intermediate DS0 TSI's in transmission systems DS-1 or above that use RBS	1.2-1.5 dB 2nd 0.6-1.2 dB 3rd	Lack of superframe alignment can cause bits to be robbed from different frames resulting in additional RBS.	Insure superframe alignment throughout DS-1 path.
Digital Loss PADs	Digital Switches	1.5-2.5 dB	0,3 or 6dB of loss inserted digitally to control echo depending on circuit length	Use adaptive hybrid balance networks to minimize echo. Do loss in analog portion of circuit.

* Equivalent SNR loss is relative to the measured 37-39 dB [8] associated with a single PCM link. TR- 57 [9] specifies the minimum SNR as 33 dB.

5.2.1 Digital To Analog Conversions (CODECs & Hybrids)

In the PSTN, nearly every connection involves at least a pair of hybrids[5] and a Digital-to-Analog (D/A) and Analog-to-Digital (A/D) conversions (i.e., CODECs).

Digital COder/DECoders (CODECs) introduce quantizing distortion. Quantizing distortion occurs due to digitization of the amplitude of a continuous electrical signal (microphone signal if VF, QAM if V.34 modem) into 255 discrete steps under Mu-law encoding. Most voice circuits today that traverse the network have only two CODECs--one in each access. The signal to noise ratio of a single PCM link, at 1004 Hz, is typically 37- 39 dB and the resulting noise level is barely audible to the human ear. However, the SNR limits the maximum achievable DCE connect rate to 33.6 kb/s with today's modem technology[6].

Remedies for CODEC distortion require limiting the number of CODECs in the circuit. One way to do this is to eliminate the CODECs associated with the ISP portion of the call. This can be accomplished by using a digital basic or primary rate ISDN interface and 4-wire digital modems in lieu of the usual 2 wire analog modem pools. ISDN allows for bits to be sent directly without

[5] Digital switches are, by definition, 4-wire, requiring a hybrid to interface to a 2-wire loop.

[6] Table 2 indicates a minimum SNR of 36 dB is required to operate at 33.6 kb/s. As will be shown later, other impairments, like digital loss pads, further reduce the SNR quickly eroding the SNR margin and limiting the modems ability to connect at the higher rates.

reconversion. The ISP multiplexer can simply re-interpret the actual Mu-law samples and converting them back to the original digital signal.

Hybrids are circuits that are used in coupling 4-wire networks (like the trunk or I/O network) to the 2-wire access network. The problem with hybrids is the difficulty of matching the impedance of the hybrid to the variable impedance of the 2-wire loop plant they connect to. Hybrid balance therefore, is a function of the loop. Impedance mismatches allow transmitted energy, in the form of talker or listener echo, to reflect back into the circuit as shown in Figure 8.

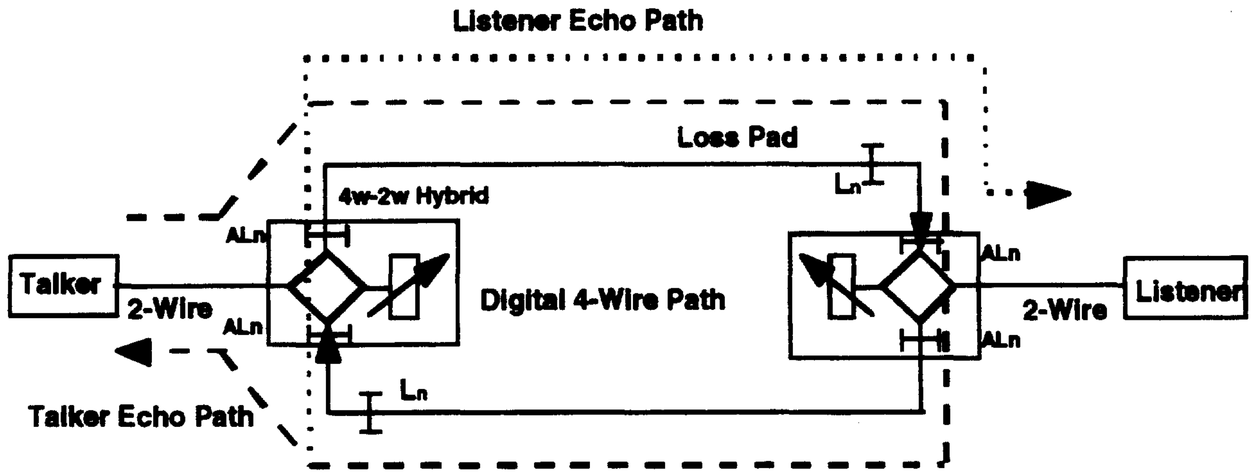

Figure 8: Talker and Listener Echo Paths

Echoes resulting from poor hybrid balance can occur in both directions. Far end talker echo is the original signal reflected back and displaced by one round trip delay. Listener echo is more difficult to cancel since it traverses the circuit 1.5 times. Listener echoes on connections with round trip delays greater than a few msec are not cancelled by most modems. To help mitigate listener and far echoes, most digital switches are capable of inserting either 0, 3 or 6 dB loss pads ("Ln" in Fig. 8) in the receive ends of the circuit, with longer circuits getting more inserted loss. Analog loss pads are indicated by ALn and are used in DLCs to adjust the gain for short loops. Older channel units typically have a fixed loss of 1 dB while newer ones have Automatic Level Compensation (ALC) and can insert as much as 5 dB of loss for shorter metallic pairs. Analog loss pads are inserted in the analog portion of the 4-wire legs (after the CODEC) and are the same for both transmit and receive directions.

Early line cards used "compromise" balance networks of fixed impedance characteristics. Fixed balance networks result in sufficient echo control for average length loops. However, fixed balance networks are not a good match for longer and loaded loops. Echoes due to poor hybrid balance on longer loops can be improved by using adaptive balance networks. These networks sense the impedance of the loop and automatically adjust to limit the amount of energy reflected back into the

4-wire portion of the circuit. Channel units with adaptive hybrid balance will improve modem performance on longer loops.

The ITU V.8 standard specifies a 1.8 to 2 second waiting period before the initial training phase of the modem. The standard has its history from old billing specifications. It seems that early in modem history, two modems could actually connect, send a burst of data and then hang-up before billing started. Newer billing technology has changed this. Where network adaptive hybrids are concerned, it would still be prudent for modems to wait the two seconds to allow the network to "calm or adapt" before modem training.

5.2.2 Robbed Bit Signaling and Superframe Alignment

Robbed Bit Signaling (RBS), which was designed into the network long before echo canceling modems were developed is another potential impairment to V.34 modem performance. RBS is a form of in-band signaling that uses the least significant bit of every sixth DS-1 frame to convey telephony signaling information. The Bellcore standard for integrated digital switch interfaces such as TR-08 and, the more recent TR-303 [10], use RBS.

Inband signaling, by definition, uses a small amount of data bandwidth to provide call processing information within an access system. This increases quantization noise, which may prevent a modem from connecting at its highest rate. On the other hand, integrated DLC systems in the loop can significantly improve signal level attenuation and attenuation distortion which improves the overall Signal-to-Noise-Ratio (SNR). Note that RBS does not alter the end user's data.

The PCM process begins with the analog signal which has been AC coupled from the loop to eliminate any DC bias. The signal is filtered and sampled and then quantized using a compressed, non-linear transfer function called Mu-law to encode the samples into one of 255 discrete levels. Signal values range from minus 127 to positive 127 and represent the various amplitude levels of the analog input signal. The middle values represent low energy levels (whispers), which are quantized in smaller steps according to the Mu-law function. RBS can become a problem at lower signal levels since the PCM SNR is already lower for these levels.

Essentially, the robbed bit appears as a low level impulse hit and results in an increase in quantizing distortion of about 2.5 dB in the decoding process. Statistically, the error occurs 50% of the time and corrupts the least significant bit of a random PCM byte one twelfth of the time. It is more likely to affect the performance of the modem as more DLC systems and independently timed DS-1 links (where superframe alignment is not maintained) are added to the connection.

Additional quantizing noise can occur because superframe integrity is not maintained through certain types of network equipment. This is especially true when signals traverse Time Slot Interchangers (TSIs). If out-of-band signaling is used (as is the case with the trunk network and Common Channel Interoffice Signaling), lack of Superframe alignment should not cause problems. If RBS is used in more than one PCM link between TSIs, bits can be robbed twice (once in each independent DS-1 leg) effectively increasing the quantizing distortion by 1.5 dB.

Experiments conducted at Lucent Technologies have shown that RBS will not, in itself, keep V.34 modems from connecting at 28.8kb/s.

5.2.3 Digital Loss Pads

Digital switches insert loss based on call routing to reduce annoying echoes generated from long circuits. The amount of loss can be applied on a trunk by trunk basis and is fixed for that trunk. Since trunks connect adjacent offices, trunks that link toll tandems can be set up to insert more loss. The odds of having a 0, 3 or 6 dB pad inserted on the call depend on the physical length of the call (i.e., within the local serving switch-0 dB, through the Interoffice network- 3 dB or through the toll network- 6 dB). Typically, if the call is "local" (as is the case with most internet calls), no loss pads will be inserted. One exception is if a call goes to overflow and is routed through a tandem switch. In this case, it is possible to have digital loss inserted into a call of short physical length.

Modems typically do not cancel listener echoes because of the long delay. Without loss built into the receive path, listener echoes could become severe enough to limit modem performance. One solution to the listener (and talker) echo problem is controlled impedance matching between the local loop and the balance network of the network hybrid. An adaptive hybrid balance can help minimize the amount of loss required in the Digital Loss pad. For example, if hybrid balances of 25 dB or better can be achieved the need for digital loss pads would disappear[7].

Digital loss, while effective for controlling echo, essentially introduces additional quantization noise. Note that loss pads inserted in the analog portion of the circuit are linear and do not introduce additional noise.

[7] The practical problem is that digital loss is selected based on dialed digits (call routing), whereas adaptive hybrids would only be present in newer DLC channel units and switch line cards (independent implementations). Toll switches would not know which connections had the newer channel units.

5.3 Other Impairments: T-Carrier BER , Synchronization, Echo Cancellers & Voice Enhancements

There are other network elements and impairments in addition to the ones discussed previously that can affect modem performance. A summary of these impairments appear in Table 5.

Table 5: Other Impairments That Can Affect Modem Performance

Network Equipment	Network Location	Equiv SNR Loss/ Occurrence	Impairment(s)	Some Remedies
Network Echo Cancellers	Satellite transmission systems. Long distance networks.	N/A	Work against echo cancellors in the modems	Disable on data calls by detecting modem tones.
T-CXR Bit Error Rates	Copper based T-Cxr systems	N/A	BERs that do not affect voice may affect data.	Use fiber transmission. Check BER on T-CXR (10^{-4} or better)
Synchronization & Timing Recovery	Bi-polar/AMI and Superframe Systems	N/A	Loss of sync due to aliasing of framing sequence and/or slips due to improper network design.	Use Extended SuperFrame. Use SONET.
Voice Enhancement	Switches and line equipment	N/A	Pre-emphasis etc., causes distortions	Disable on data calls by detecting modem tones.

Digital carrier systems have undergone an evolution since their incorporation into the network in the early 1960's. Early "T" Carrier systems used straight bi-polar (alternate marks inversion) techniques as the line code. The line repeaters used a timing recovery circuit that lost synchronization when more than 14 consecutive zeros were transmitted. The early solution was the use of Zero Code Suppression (ZCS), in which the least significant bit of an all zeroes byte is forced to a one, without subsequent correction. As data applications became more prevalent, these systems degraded quality unacceptably, and some other method for eliminating long strings of zeroes was required. B8ZS (Bi-polar 8 Zero Substitution) replaced the ZCS method to improve data quality. B8ZS is used in newer SuperFrame (SF & Extended SF) systems (these systems improve synchronization and provide a low bit rate channel for system maintenance). T-cxr systems go into an alarm condition when the BER exceeds $1X10^{-3}$ however, most carrier systems operate at much lower error rates.

T-cxr digital errors translate into the analog domain as impulse hits with an amplitude dependent on the errored bit. Errors in the least and most significant bits translate into small and large impulse hits respectively. Thus, on well maintained carrier systems, V.34 modems operate essentially error free. Older carrier systems with marginal Bit Error Rates (BER), that may still be acceptable for voice, can also cause poor modem performance.

Loss of synchronization can occur between networks and even within a single carrier's network. Framing slips are deadly to data and may cause the modems to retrain. Slips are usually due to improper equipment set-up. For example, local timing at one or both terminals at the ends of a transmission system can cause slips. Losses of synchronization can usually be detected since they occur at regular intervals. To prevent these types of problems, careful deployment of network equipment is required.

Network Echo Cancelers (NECs) may also prevent modems from operating. Fortunately, modern NECs can detect modem training tones and disable themselves for the duration of the call. Since most echo Cancelers in use today are of the modern variety and internet access connections tend to be short, NECs should not present a problem to end users.

Any of these "other digital impairments" can result in modem errors or loss of modem data. And, as already discussed, excessive retransmissions have a deleterious impact on modem throughput.

The important aspect of digital impairments is that they affect analog signals traversing a digital network. V.34 modems have pushed the envelope of analog transmission within the confines of standard telephony practices. These impairments generally do not occur in all-digital services such as Digital Data Service (DDS) and ISDN. These services are predicated on completely digital transmission from end-to-end thus never have to deal with intermediate CODECs, hybrids and access plant conditioned for voice. In fact, changing the assumptions relative to the channel characteristics over which modems operate is how the next generation of modems will be designed.

5.4 Impairment Summary

Analog and digital impairments affect the Signal-to-Noise Ratio and the attenuation loss seen by the receiver on the connection. Both must be considered in estimating modem connect rates for any given PSTN connection. While the process of signal level and SNR erosion can be illustrated, experience has shown that actual results are difficult to predict because of the number of variables that need to be considered. The example shown in figure 9 is intended to demonstrate how quickly the design limits for signal level and SNR are exceeded given just a few of the variables that require consideration. In reality, it is difficult to predict how a modem will perform on a given connection without experimentation.

In this example, the end user connects to the DLC system through a 10 kft non-loaded, 26 gauge, metallic pair. The DLC system is a Universal Digital Loop Carrier System. The circuit is converted to a 2-wire analog signal and is connected to the line card of the digital switch. Finally, the ISP access connection is a 26 gauge, 10 kft, non-loaded loop.

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High Speed Modem PerformanceOverThe PSTN

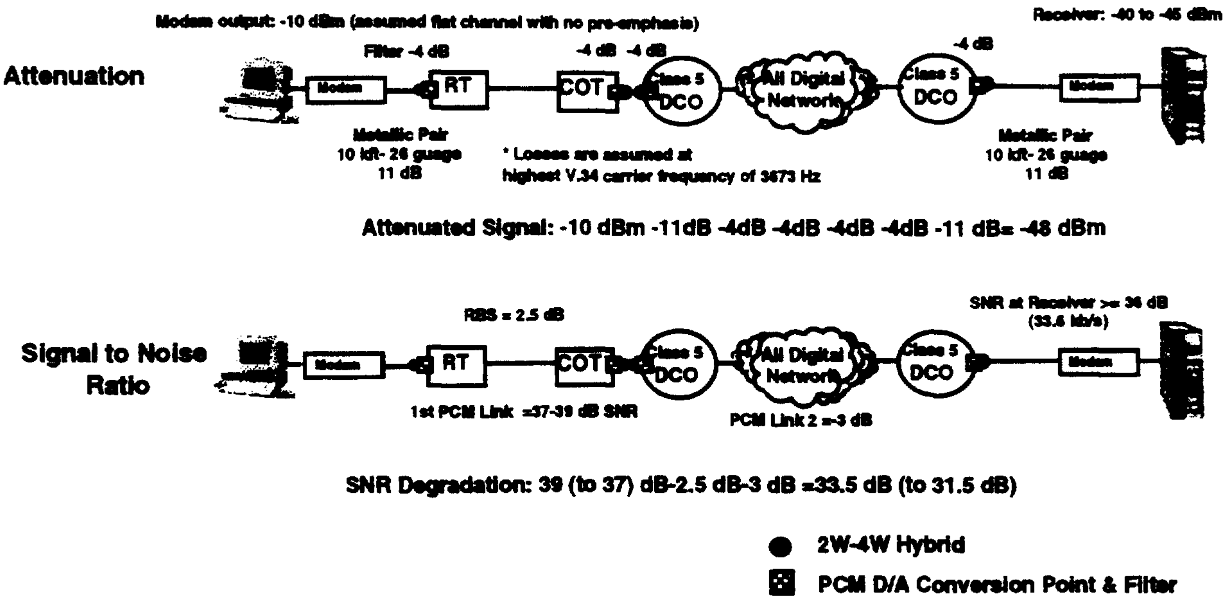

Figure 9: Example Impairment Calculations at 3700Hz (for illustration only)

Recall that most V.34 modems require a -40 to -45 dBm receive signal level. In the example above, the attenuation values for frequencies associated with the upper Nyquist frequencies were used to estimate the loss of the connection. If the signal starts at -10 dBm, after 4 filters and two metallic loops, a signal at 3700 Hz would be around -48 dBm (using the optimistic receive value of -45 dB) and may be undetectable by the modem. In addition, the SNR required (from Table 2) to achieve a 33.6 kb/s connect rate is approximately 36 dB. The SNR degradation from 2 PCM links and RBS reduces the signal-to-noise margin of the system from 37-39 dB to between 31.5-33.5 dB which is insufficient to achieve 33.6 kb/s DCE connect rates[8]. Due to the impairments, the modem is likely to drop down to a lower symbol rate and DCE connect rate. The above example assumes no benefit from precoding and pre-emphasis algorithms in the modem.

Connections such as the one used in this example are being described for purposes of unbundling the loop for local access competition. As the example shows, impairments due to the PCM process and signal level impairments should not prevent a modem from connecting however, it could prevent connections at the higher V.34 rates.

[8] Recall that the actual minimum PSTN requirement for SNR for a set of CODECs is only 33 dB. The actual SNR the modem must operate with could be even lower than cited in this example.

V.34 modems are extremely complex and can perform radically different when presented with different types of impairments. Nothing short of experiments can predict their actual performance on a given PSTN connection

5.5 Statistical Characterization of Network Connection Combinations

In terms of modem performance, there are four dominant areas of variability that need to be considered: the user modem, the ISP modem and the two respective access networks. The user modem and user access plant configurations cannot be controlled. They are highly variable but remain constant from call to call. ISP connections are the most controllable portion of the link but specific equipment seized for a call can vary from call to call (and especially from ISP to ISP). For example, different vintages and brands of modems may be mixed in the same modem pool. This section examines the variability of each of these four circuit components.

5.5.1 ISP Access

Figure 10 illustrates three ways (A,B & C) ISP POPs can be connected to the PSTN.

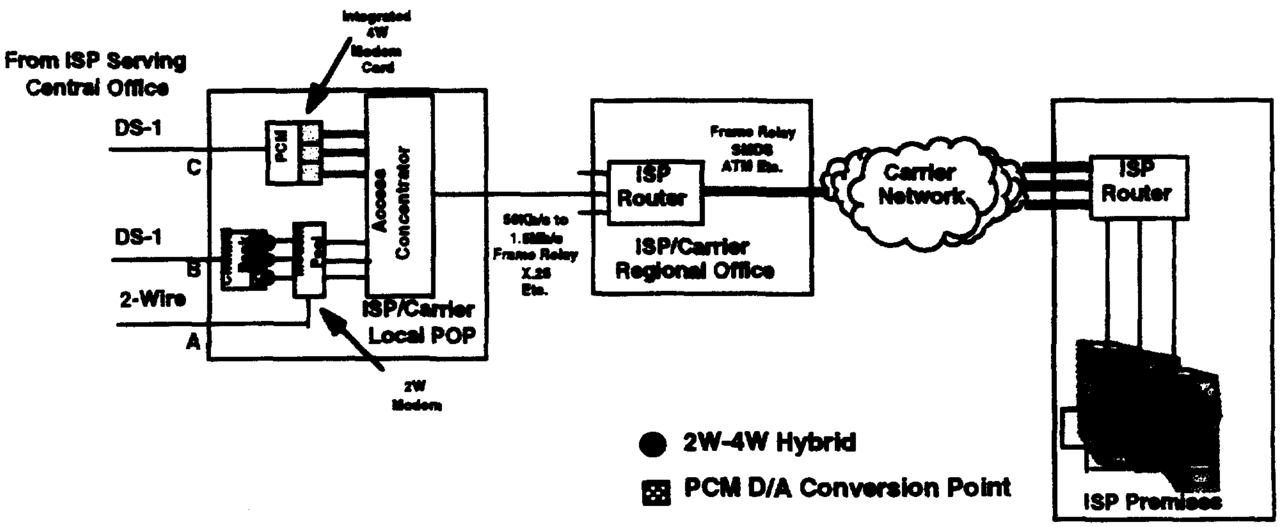

Figure 10: Three Types of ISP Network Access

The simplest way is to reverse the user's typical access process by decoding the digital signal and remodulating it onto a 2-wire telephone line (1MB) and connecting it to an analog modem at the ISP POP. A Multi-Line Hunt Group (MLHG) is established with many lines connecting to a modem pool via a guide number the user dials. Currently, this is the most common form of ISP access but also the most impairment laden end to end because two analog loops are always involved in the connection on either side of an all digital network (note: configuration A would be ideal for a connection through an analog switch, where the user and ISP switches were one and the same, since no CODECs or hybrids would be involved).

Configuration B shows a modification of the MLHG process. Here, the D/A conversion and hybrid are pushed into the ISP POP. While the number of D/A conversions and hybrids remains the same as in A, this configuration avoids the second appearance of a local loop in the circuit. This effectively eliminates any attenuation distortion due to the link between the CO and POP. Usually, a DLC remote terminal set up in TR-08 or TR-303 integrated mode is used in this application to avoid the conversion back to 2-wire. This configuration can also be taken off of the trunk side of the ISP's serving switch via the Digital Trunk Frame (DTF) and served via a digital channel bank. This obviates the need for a DLC remote terminal but may lack remote testing capabilities which will encumber trouble shooting of the ISP access leg.

The third access mechanism is similar to B in that it eliminates the second loop and its transmission attenuation. In this case, a Primary or Basic rate ISDN line is used (a subscriber DS-1 could be used also). This has two advantages: 1.) out-of-band signaling eliminates the quantizing distortion caused by RBS and 2.) the ISP can use 4-wire digital modems thus avoiding two additional hybrids (one in the channel bank and the second in the 2-wire modem) and a CODEC. This configuration introduces no impairments in the ISP access link and has the theoretical transmission limit of 64 kb/s per user. It is also upwardly provisionable to ISDN for high speed users. In short, it is exactly the data application that Primary Rate ISDN was intended for.

The diagram shows ISP access concentrators connected to a regional router which in turn is connected through a carrier network to the ISP's servers. This network represents a large ISP network. Smaller ISPs may connect directly to their servers without going through intermediate routers or networks. Generally, traffic is backhauled to one, or a few, data center locations since management of the servers in a distributed environment is difficult and potentially expensive.

5.5.2 User Access

The second area that needs to be factored into modem performance is the design and variability of the telephone access plant. With 100 years of evolution behind it, the access network is probably the most diverse, least engineered and the most difficult to change. When V.34 specifications were developed, a model of the access plant was required to understand how various technical alternatives would play against the loop population. Unfortunately, the loop population was last characterized in a survey performed by Bellcore using data collected in 1983. Many changes have occurred since then.

TIA TSB37-A [3] specifies seven loop models based on the 1983 Bellcore work. Figure 11 illustrates all seven models with the corresponding original estimates of the percentage of loops that fall within a particular category.

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High Speed Modem Performance Over The PSTN

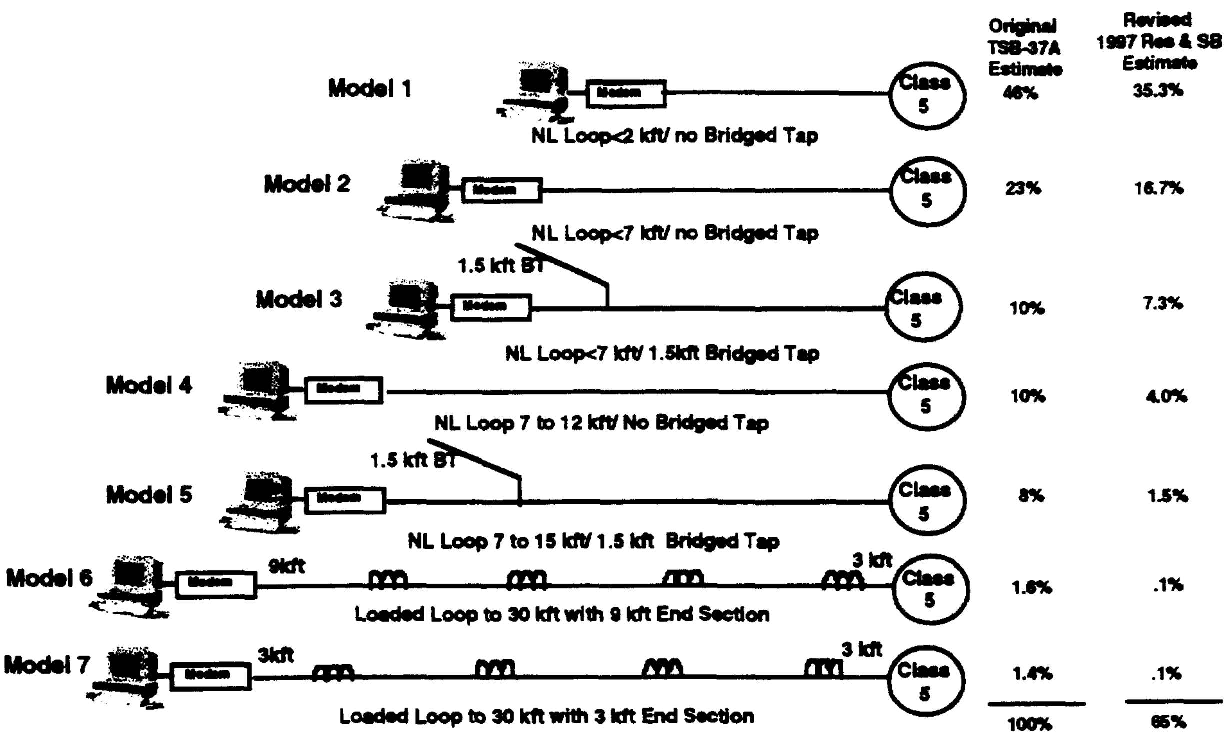

Figure 11: TSB37-A Loop Population

The seven models of the study included no Digital Loop Carrier statistics. In order to assess the impact of the loop on modem performance for Residential and Small Buisness (Res & SB) users, it is useful to extrapolate the data to include DLC models. The result of this "reassessment" is shown under the column of Revised 1997 Res & SB Estimate (in figures 11 and12) and should be considered an educated guess. Since the primary users of modems are, and will continue to be, Residential and Small Businesses, it was necessary to parse the original data (which included all types of metallic loops) to only these two segments. Keep in mind that large business users typically do not dial into the internet via the PSTN.

Four additional models were developed based on an understanding of the applications that Outside Plant Engineers use to deploy DLC systems. They appear in figure 13 and include both Universal and Integrated DLC applications. DLC systems deployed in the OSP always have a metallic pair between the Remote Terminal and the end user's premises. CSA design rules stipulate that the metallic portion of the loop be no longer than 12 kft, have minimal bridged tap and contain no load coils. The metallic portion of DLC loops then look very much like models 1-4 in figure 12.

The reassessment was performed by using current knowledge of how long a DLC configuration/technology has been available and how it was used. The assumptions were based on estimates that peg Residential and Small Business loops at about 80% of the loops in the U.S.

This leads to a percentage of DLC penetration of about 15% within the Res & SB sectors (near 20% overall). The penetration of DLC will vary from LEC to LEC. Since DLC systems began as Universal configurations and only proved in for longer routes, it is safe to assume that more of the longer loops were replaced over time with Universal DLC systems. Similar assumptions can be made regarding the general loop population over time, namely that DLC has become more widely deployed for ever decreasing route lengths. The DLC reallocation was performed under the following assumptions:

Model 1	2% reallocated to DLC
Model 2	10% reallocated to DLC
Model 3	10% reallocated to DLC
Model 4	50% reallocated to DLC
Model 5	75% reallocated to DLC
Model 6	90% reallocated to DLC
Model 7	90% reallocated to DLC

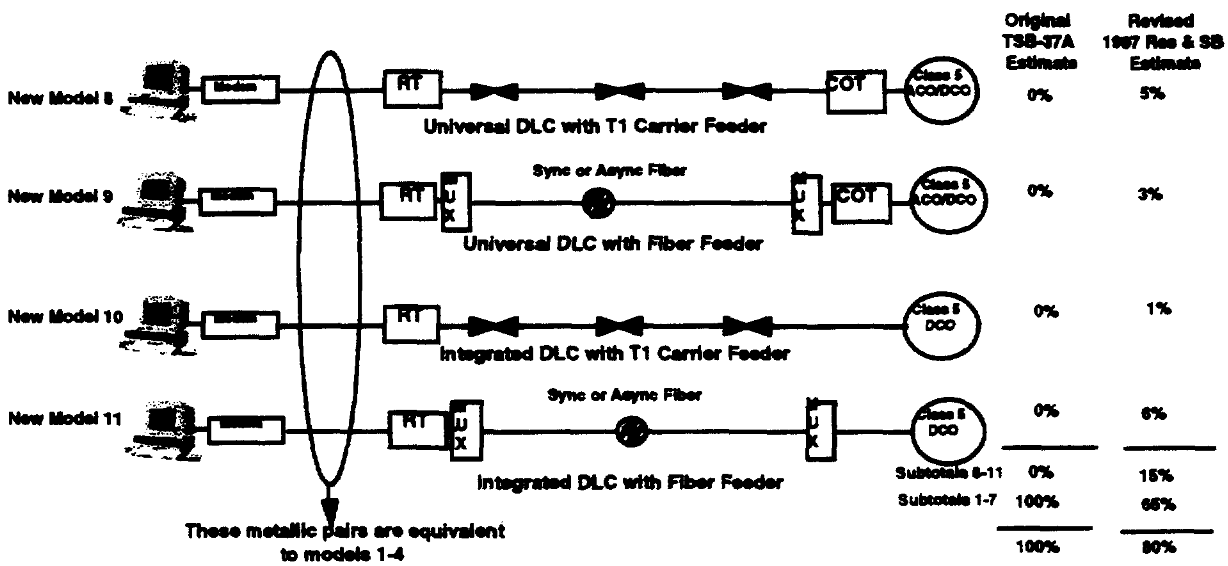

Figure 12: DLC Configurations and Summary of End User Access Percentages

These statistics indicate that about 12% ((2% long loops+ 8% DLC loops)/80%) of the loop population may experience user access-based degradation in V.34 modem performance at 28.8kb/s, in which the modem would connect at a lower rate. This includes Universal DLC connections that contain one PCM link (perhaps three PCM Links end to end for a UDLC to UDLC connection). These Universal DLC circuits, representing about 10% (8%/80%) of the total Res & SB population, can still achieve V.34 (16.8 kb/s) or higher DCE connect rates depending on the individual circuit and type of ISP connection. Performance for these end users will depend on the length and condition of the metallic pairs connected to the DLC and how robust the modem receivers are.

Digital PBXs and key systems on customer premises can also introduce an additional PCM link into a connection if they are connected to the PSTN via analog trunks or lines. The probability of encountering this situation is unknown but a solution is to always use a directly connected line for high speed modems and fax machines.

Even though these are gross estimates, they should be useful in understanding the probability of voice band modem users encountering certain impairments as they access an ISP. While there are many impairments presented to the analog modem by digital systems, telephone companies have eliminated most of the longer and more troublesome analog copper pairs (models 5,6 &7 or (1.5+.1+.1)/.8= 2%). This has improved overall modem performance across the user access network.

5.5.3 Modem Variability

The third area of potential variability is in the modems themselves. While most modems work consistently over short copper pairs, their performance over long loops, coupled with multiple hybrids and PCM links, varies considerably. It is assumed, without knowing the specifics of how the modem was designed, that modems were manufactured with differing optimization algorithms. For example, one modem might try to connect at the highest possible DCE connect rate even though errors cause excessive retransmissions while another might choose to connect at a lower rate in order to operate error free. The result of modem testing, if nothing else, corroborates this.

Laboratory tests were conducted using a range of loop configurations similar to those described in the previous section. Modems A & B, from the same manufacturer, were paired to perform the tests. Test were done in both directions and are arranged by loop length. Figures 13 and 14 show modem connect rates for various digital access arrangements that had from 0 to 3 PCM links. Within each digital access arrangement, a number of metallic pair configurations were tested, which leads to the scatter within each vertical band. Figure 13 shows the connect rate of a modem that performed better under most conditions. Figure 14 illustrates a modem that performed well on average but poorer than the modem in Figure 13. Note that, in general, a single PCM link did not prevent modems from achieving a connect rate of 28.8 kb/s[9], especially for configurations with shorter metallic end sections.

[9] The modems used in the tests were manufactured prior to the adoption of the higher, optional, V.34 rates, thus the maximum DCE connect rate was 28.8 kb/s.

Lucent Technologies

High Speed Modem PerformanceOverThe PSTN

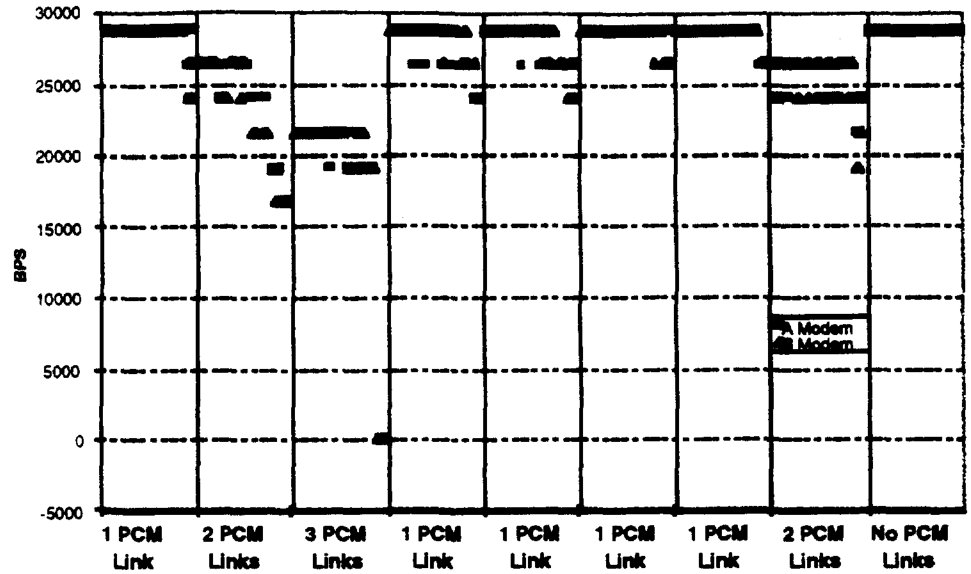

Figure 13: Better Quality Modem

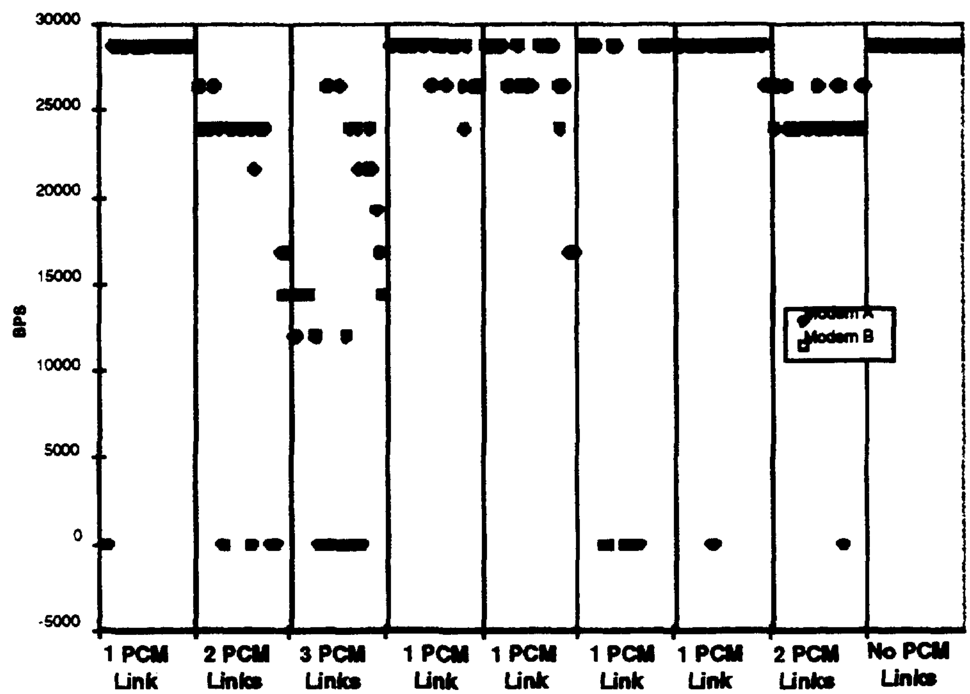

Figure 14: Poorer Quality Modem

In general, V.34 modems performed as expected. Variations between modems were probably due to variations in receiver design. Recall that standards specify the transmitter output but not the receiver design. Most of the lower scores in these tests were caused by access arrangements that occur infrequently.

6. Mu-law Modems

Just around the corner is a new breed of modem designed to work at DCE rates up to 56kb/s. These modems will be designed with a very different set of operating assumptions. To date, modems have been designed with the starting assumption of analog loops at the end user and ISP ends inter-connected by a digital inter-exchange network. New 56kb/s designs will operate under the assumption of a single network CODEC, allowing an analog end user access but requiring a digital, 4-wire, ISP connection. Changing the operating assumptions around which modems are designed allows for new transmission schemes. While it is unclear, at this point, which user access models will work at 56kb/s, Universal DLC configurations can be ruled out because of multiple CODECs.

New 56kb/s modems may tend to be asymmetric. Current proposals use 56kb/s toward the customer and V.34 rates upstream since equalization is more difficult at the user end. The operating premise of these new modems is that they are synchronized to a single PCM CODEC in the PSTN. In other words, the system simply assumes Mu-law quantization. Figure 15 is an example of a possible implementation of a 56kb/s modem.

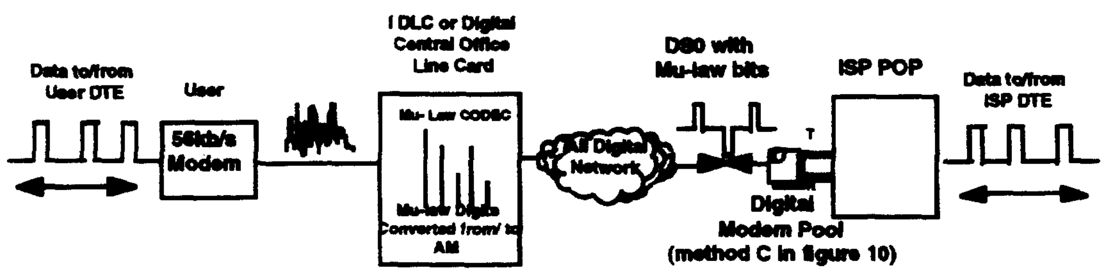

Figure 15: Example of 56kb/s Modem

The example shows a serial bit stream from an ISP server being introduced to a digital modem. The network transports the user data as Mu-law bits. The CODEC in the line card facing the user accurately reproduces an analog signal, amplitude modulated by the original Mu-law bits. The signal is sent out toward the end user over the local loop and is equalized and processed in the user 56kb/s modem receiver. The user's receiver must be synchronized to the network CODEC's 8kHz sampling clock. The user's transmitter, so synchronized, sends data toward the ISP by reversing this process. The user modem may also default back to V.34 for upstream.

To further understand how these systems may be able to transmit at rates up to 56 kb/s, it is important to understand the Mu-law encoding process. Figure 16 illustrates the non-linear nature of Mu-law encoding which is important to 56kb/s modem design. Eight bits generated 8000 times per second, yields a 64kb/s transmission rate- the rate of a single DS0 channel.

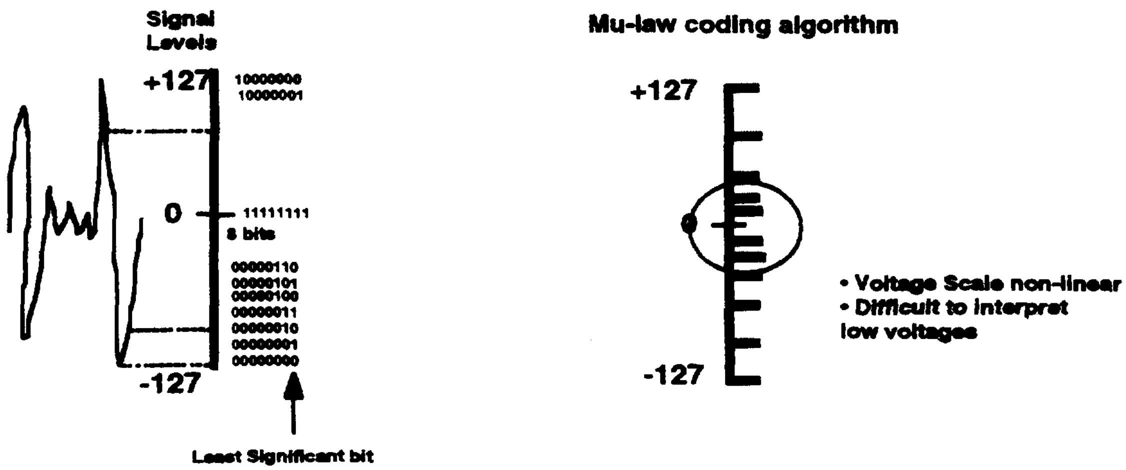

Figure 16: Mu-law Encoding

In reality, because of the circuit noise and attenuation distortion, this rate will be difficult to achieve. Since most loops (except for special services like DDS and ISDN circuits) are not "designed" to specific transmission parameters, 56kb/s modems assume that near zero voltage levels will be indistinguishable. Table 6 examines the impact on transmission rates when less than 256 levels can be detected.

Voltage Levels	Bits per PAM Sample	DCE Rate bps
2	1	8000
4	2	16,000
8	3	24,000
16	4	32,000
32	5	40,000
64	6	48,000
128	7	56,000
256	8	64,000

Table 6: Quantizing Levels Required to Achieve Various Bit Rates

The interesting thing to note, is that only half of the available levels in the Mu-law encoding algorithm need to be detected to achieve a DCE connect rate of 56kb/s. In reality, levels around zero will probably not be used at all, every other voltage level will be used in medium energy ranges and potentially all levels in the high energy ranges will be detected by the receiver in order to achieve the 128 levels sufficient to attain 56kb/s.

Since 56kb/s modems are not yet in wide scale use and have not been standardized, not much is known about how the modems will respond to various impairments. Since the signal is amplitude modulated, attenuation distortion associated with longer loops and load coils will have a significant effect on these systems. Since the number of CODECs, and corresponding hybrids are, by definition, limited to one, echo should be less of a problem. Even if near zero Mu-law levels are not used, connect rates of 56 kb/s should be possible with this approach.

7. Wrap-up

The advent of the internet and an industry direction toward "telecommuting" has changed the way information is exchanged around the world. The internet has spurred interest in a new generation of analog voice band modems that permit high speed access via the PSTN. At the highest connect rates, a V.34 modem operates on the passband fringes of the telephone network in its current state. Meanwhile, the network continues to evolve. Variabilities in (non-designed) loops across the PSTN and modem quality do not permit guaranteed modem performance. The issue is not so much whether the modem will connect but rather at what rate it will connect.

Modem performance is limited by four dominant factors: analog impairments in the user's access, digital impairments from the PCM process, the ISP access type, and individual modem variability. Each taken on its own is insufficient to understand the variability in modem performance. A keen understanding of how each piece plays a role is essential in understanding modem connect rates and data throughput. Connect rates of 19.2 kb/s should be considered the minimum rate achievable on all but the longest, most troublesome, loops.

Modem performance improvements can be gained by:

- upgrading ISP connection from analog 2-wire modems to digital 4-wire modems,
- improving loop attenution and attenuation distortion by using DLC (or new broadband access systems) to shorten the effective length of the metallic pair,
- converting UDLC systems to IDLC whenever a digital switch is introduced into the serving CO and
- using adaptive balance networks in channel units to reduce echo.

All of the impairments discussed here can be improved going forward. Retrofitting installed equipment may be cost prohibitive.

In the future, modem manufacturers and network designers need to coordinate how each piece is designed to insure a balance between voice quality and data throughput. For example, modem manufacturers should observe the 2 second network quieting period before training. Similarly, network designers should avoid unnecessary voice spectrum shaping functions in POTS circuits. Modem manufacturers need to educate their customers with respect to modem performance on the PSTN.

The network today consists of different technologies each designed to meet the criteria of its day. However, the network is optimized for voice communications. This makes it impossible to guarantee modem performance on non-designed telephone circuits. It is anticipated that a high percentage of Internet users will access the Internet via V.34 and Mu-law based modems in the future. These rates push the limits of today's PSTN and often create misunderstandings relative to modem performance. The only realistic way to guarantee high speed data communications is to move to a clear channel, all digital access like ISDN on <u>both</u> user and ISP access facilities.

8. GLOSSARY

ASCII- American Standard Code for Information Interchange
B8ZS- Bi-polar 8 Zero Substitution
BER- Bit Error Rate
BLER- Block Error Rate
CAP- Competitive Access Provider
CLEC- Competitive Local Exchange Carrier
CO- (Telephone) Central Office
CODEC- Coder/Decoder
COT- Central Office Terminal (for Universal DLC)
CRC- Cyclical Redundancy Check
CSA- Carrier Serving Area
D/A- Digital to Analog
DCE- Data Connection Equipment
DDS- Digital Data Service
DLC- Digital Loop Carrier System
DPSK- Differential Phase Shift Keying
DTE- Data Terminating Equipment
DTF- Digital Trunk Frame
ESF- Extended SuperFrame
FDM- Frequency Division Multiplexing
Hz- Hertz
I/O- Interoffice Network
IDLC- Integrated Digital Loop Carrier System
ISDN- Integrated Services Digital Network
ISP- Internet Service Provider
ITU- International Telecommunications Union
LEC- Local Exchange Carrier
MDF- Main Distributing Frame
MLHG- Multi- Line Hunt Group
NEC- Network Echo Canceler
OSP- OutSide Plant
PCM- Pulse Code Modulation
POP- Point Of Presence
POTS- Plain Old Telephone Service
PSTN- Public Switched Telephone Network
QAM- Quadrature Amplitude Modulation
RBS- Robbed Bit Signaling
RMS- Root Mean Squared
SAI- Serving Area Interface
SF- SuperFrame
SNR- Signal to Noise Ratio
TSI- Time Slot Interchanger
ZCS- Zero Code Suppression

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